In the Specification:

Please replace paragraph [0004] of the Published Application with the following rewritten paragraph:

In order to be able to transform an existing telephone network which conveys voice signals over twisted pair lines into a multi-media network it is known to provide for ~~tis~~ this purpose as asymmetric digital subscriber line (ADSL). An ADSL is a point-to-~~pint~~ point connected circuit which affords each subscriber to the network with a high-speed communication link that in addition to the usual telephone services affords many other services such as video-on-demand, conference video phone communication and high-definition TV as well as a full range of Internet services.

Please replace paragraph [0006] of the Published Application with the following rewritten paragraph:

Channel A: a high-seed downstream channel running ~~form~~ from the central office of the network to an end user.

Please replace paragraph [0010] of the Published Application with the following rewritten paragraph:

Currently available are two ADSL systems which comply with current regulatory telephone standards, namely: a split-type ~~ASL~~ ADSL and a splitterless type. In an ADSL of the split-type, also known as a Full-Rate type, the voice signals which are produced concurrently with the digital data signals are split therefrom and conveyed to the central station of the network over a twisted ~~par~~ pair cable, whereas the data signals are conveyed over another twisted pair cable.

Please replace paragraph [0012] of the Published Application with the following rewritten paragraph:

~~As~~ An ADSL of the split type cannot be installed ~~i~~ in residences or offices having the usual two-wire internal telephone line wiring. It is necessary in order to install a split-type ADSL in ~~this~~ these facilities, to provide an additional two wire line running from the modem

to the splitter. Such rewiring substantially raises the expenses incurred in installing an ADSL system.

Please replace paragraph [0014] of the Published Application with the following rewritten paragraph:

Existing ADSL systems of the "splitterless" type which function to convey both digital data and voice signals simultaneously over a single twisted pair telephone cable include either a POTS splitter to separate the voice ~~form~~ from the data or a POTS Line Card for this purpose. These are relatively expensive components.

Please replace paragraph [0019] of the Published Application with the following rewritten paragraph:

More particularly, an object of this invention is to provide a system of the above type which includes a ~~splinterless~~ splitterless asymmetric digital subscriber line (ADSL) that obviates the drawbacks of prior art ~~a~~ ADSL ~~system~~systems.

Please replace paragraph [0035] of the Published Application with the following rewritten paragraph:

FIG. 10A is a flow chart illustrating the sequence of processing events in data processing in ~~he~~ the transmitter;

Please replace paragraph [0045] of the Published Application with the following rewritten paragraph:

FIG. 2 is a flow chart of the processing steps carried out in an ATU (ADSL Transceiver Unit) ~~reference~~ referenced in ITU (International Telecommunications Union) recommendation G.992.2, referring to the splitterless ADSL.

Please replace paragraph [0046] of the Published Application with the following rewritten paragraph:

Data 153 is processed in ATM Cell Formation step 151 by an interface port resulting in a sequence of ATM (Synchronous Transfer Mode) cells. In step 155 the cells are RS (Reed--Solomon) encoded and scrambled. The ADSL system employs FEC (Forward Error Correction) based on RS encoding to reduce the effect of the impulse noise. In step 157 an

interleaver mixes data bits to protect the encoded data cells from impulse noise. In step 159 tone ordering is calculated for the interleaved encoded data and the data is distributed ~~amount~~ among 128 tones (carriers) of multitone line signal. In step 161, modulation parameters are calculated by a constellation encoder and a gain scaler for each carrier. In step 163 the modulation parameters of all carriers are transformed by an IDFT (Inverse Discrete Fourier Transformation) processing to produce digital samples of DMT (Discrete MultiTone) signals. In step 165 the digital samples are written into an output buffer. In step 167 a DTA (Digital To Analog) converter transforms the digital samples to analog DMT line signal.

Please replace paragraph [0047] of the Published Application with the following rewritten paragraph:

The ADSL is an adaptive system. During an initialization ~~Communication~~ communication phase, an ADSL system measures the SNR (signal-to-noise ratio) for each carrier and defines the number of bits that may be loaded on the carrier.

Please replace paragraph [0049] of the Published Application with the following rewritten paragraph:

As previously pointed out, an existing ADSL system of the type shown in FIG. 1 has several drawbacks. The most serious of which is that the system provides the telephone subscriber with only a single base band voice channel which for many subscribers is inadequate. ~~And the~~The inclusion of a POTS Line Card in this prior art system adds substantially to its cost.

Please replace paragraph [0056] of the Published Application with the following rewritten paragraph:

Referring to FIG. 5 shown therein is a time base diagram illustrating the correlation between the PCM encoded voice signal and the DMT frames. Because the sampling rate of the PCM encoder is 8 kHz and the DMT-ADSL frames shown in row A operate as a frequency of 4 kHz, it is necessary to use two upstream VCs and two downstream VCs for transmitting one telephone channel. In row B, voice signal samples 2 and 3 are assigned to ~~tow~~ two different VCs in row C1 and row C2 which correspond exactly to the DMT-ADSL

frames in row A. The PCM encoder is therefore synchronized with the DMT-ADSL frame rate.

Please replace paragraph [0059] of the Published Application with the following rewritten paragraph:

As can be seen in FIG. 6B, the more significant bits of a PCM word correspond to the more significant bits of real and imaginary components of the respective QAM vector. Errors may be produced by channel noise only in the less significant bits of the PCM words because of the short distance between consecutive QAM vectors. As a result, errors in low significant bits of PCM words produce only small additional noise in the voice signal. As a consequence, the present invention promotes a high quality voice signal transmission over an ADSL system without implementing error correction coding and without significant delay. An ADSL communication system ~~i~~ in accordance with the invention may be extended to effect simultaneous transmission of several voice channels.

Please replace paragraph [0060] of the Published Application with the following rewritten paragraph:

The Basic System: FIG. 7 illustrates schematically a communication system 201 in accordance with a preferred embodiment of the present invention. A subscriber premise 103 is coupled to the CO (central office) 109 of a telephone network by a twisted wire pair telephone cable 107. At the subscriber's premise 103, the twisted ~~wrie~~ wire pair 107 is connected to ATU-R transceiver unit 205. A fax machine 121 and a telephone set 123 are connected to a voice interface port 203 of ATU-R 205, using for this purpose internal telephone lines 117. A PC ~~125~~ (personal computer) 125 is connected to a digital interface port 204 of ATU-R 205 by an Ethernet cable 124.

Please replace paragraph [0061] of the Published Application with the following rewritten paragraph:

Central ~~Station~~ office 109 contains an ATU-C transceiver unit 211, a data switch 135, a telephone switch 137, a data network 115, and a telephone network 113. A subscriber twisted pair cable 107 is coupled directly to ATU-C 211 and ~~carrier~~ carries data and voice on (DMT) discrete multi-tone line carriers. Data signals flow from data interface port 209 of

ATU-C 211 to the data switch 135. Voice signals flow from voice interface port 207 of ATU-C 211 to telephone switch 137. Telephone switch 137 is coupled to telephone network 113 whereas data switch 135 is coupled to data network 115.

Please replace paragraph [0067] of the Published Application with the following rewritten paragraph:

FIG. 9 illustrates the sequence of processing ~~event~~ events which a voice signal that is one of several incoming signals undergoes in the ATU transmitter, starting with a voice signal 251A flowing into an individual interface port. In step 252, the signal is amplified and filtered by one of the interface ports available. In step 257 the voice signal is transformed by PCM encoding into a 64 kbit/sec sequence of 8-bit PCM words, the sampling rate of the PCM coder is 8 kHz using for this purpose standard A-Law or μ-Law coding, much the same as is used by the PCM telephone systems ANSI T1 or E1.

Please replace paragraph [0069] of the Published Application with the following rewritten paragraph:

Data communication over silent voice carriers: FIG. 10A illustrates the sequence of processing events involved in data processing within the ATU transmitter in accordance with another embodiment of the present invention. Data 153 is processed in step 151 by an interface port resulting in a sequence of ATM cells. In step 155 the cells are scrambled and RS encoded. In step 157 an interleaver mixes data bits to protect the encoded blocks of data from impulse noise. In step 380, the interleaved data ~~steam~~ stream is distributed between "data carriers" and silent VCs.

Please replace paragraph [0071] of the Published Application with the following rewritten paragraph:

A RS encoder calculates parity bytes for PCM words of active voice channels and puts these parity bytes on additional "voice carrier". A VCs constellation encoder and gain-scaler transforms each 8-bit PCM word into one 8-bit QAM symbol and provides fixed 8-bit loading on each "voice carrier". The ~~Sampling~~ sampling rate of each PCM coder is synchronized with the frame rate of the DMT line signal.

Please replace paragraph [0073] of the Published Application with the following rewritten paragraph:

Incorporating a digital voice channel of the CO: An ATU-C transmitter in accordance with the invention is well adapted to incorporate the electronic communication equipment of the CO, such as the PCM telephone switch (frame relay). According to a preferred embodiment of the invention, a stream of PCM telephone words of the CO is readily processed and communicated through the ADSL system. Data is processed in the same way as in example 1. FIG. 11 to which reference is now made, illustrates schematically the incorporation of a PCM digital telephone signal ~~form~~ from a frame relay. A telephone signal comes in ~~form~~ from frame relay 282 in the form of a 64 kbit/s PCM stream. It is sent to an input of a PCM interface 283 that synchronizes the DMT signal frames with the 8-bit PCM words. The 64 kb/s PCM stream is then distributed between two VCs or DMT signal as in a previous example. To synchronize the PCM stream with the DMT signal frames, a main 8-kHz clock 285 of the frame relay is connected to the synchronizer 255 and to the PCM interface 283. The synchronizer 255 is also connected to the ADSL 400 to synchronize between the data and the T1 voice source.

Please replace paragraph [0076] as amended in the Amendment of February 11, 2003 with the following rewritten paragraph:

Incorporation of several voice channels at a subscriber premises: Data is processed and transmitted in an ATU-R in the same way as described in the first example. Referring now to FIG. 13, it will be seen that voice channel 251A is connected to a voice interface port 253A which is one of several identical ports, where the necessary amplifying and filtering is performed. A PCM encoder 257A is connected to the respective voice interface ~~ports~~ port 253A. Each PCM on the encoders has a sampling rate of 8 kHz and transforms an analog voice signal into a 64-kbit/sec sequence of 8-bit PCM words. The PCM coders use standard A-Law or .mu.-Law coding, which is the same one used in PCM telephone systems T1 or E1.